DEFENSE INFORMATION SYSTEMS AGENCY
P. O. BOX 4502
ARLINGTON, VIRGINIA 22204-4502

IN REPLY REFER TO: Joint Interoperability Test Command (JTE) **6 Nov 08**

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Extension of the Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS) 1000M Cabinet and CS1000M Chassis (including Voice over Internet Protocol [VoIP]) and DSN Option 11C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages

References: (a) DoD Directive 4630.5, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004
(b) CJCSI 6212.01D, "Interoperability and Supportability of Information Technology and National Security Systems," 8 March 2006
(c) through (g), see Enclosure

1. References (a) and (b) establish the Defense Information Systems Agency (DISA), Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification.

2. The Nortel DSN CS1000M Cabinet Digital Switching System with Software Release 4.5w and Product Enhancements (including VoIP) is hereinafter referred to as the System Under Test (SUT). The SUT met all of its critical interoperability requirements and is certified as interoperable for joint use within the DSN. The SUT is certified for VoIP specifically with certified Assured Services Voice Application Local Area Networks (ASVALANs) posted on the Unified Capabilities (UC) Approved Product List (APL). One of the optional requirements, Call Forwarding Variable processing for precedence calls, did not meet the specifications and therefore is not certified for use in the DSN. The interoperability test summary and the rest of the exceptions that were identified during testing are listed in Table 1. The DSN CS1000M Chassis employs the same software and trunk/line card hardware as the Nortel DSN CS1000M Cabinet. Analysis by JITC determined that the DSN CS1000M Chassis is functionally identical to the DSN CS1000M Cabinet for interoperability certification purposes, and it is also certified for joint use within the DSN. The DSN CS1000M Cabinet and DSN CS1000M Chassis without VoIP are referred to and marketed within Department of Defense (DoD) as the Nortel DSN 11C Cabinet and DSN 11C Chassis, respectively. Except for the absence of VoIP capability, these are functionally identical to CS1000M models for interoperability certification purposes, and are also certified for joint use within the DSN. The listed test discrepancies shown in the SUT Interoperability Test Summary have an overall minor operational impact. The SUT was tested and met the critical interoperability requirements for the following DSN switch types: Private Branch Exchange (PBX) 1, and PBX 2. No other configurations, features, or functions, except

those cited within this report, are certified by the JITC, or authorized by the Program Management Office for use within the DSN. This certification expires upon changes that could affect interoperability, but no later than three years from the date of the original memorandum (7 March 2007).

3. The extension of this certification is based upon a desktop review. The original certification is based on interoperability testing conducted by JITC and a review of the vendor's Letters of Compliance (LoC). Testing was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 25 July through 1 September 2006. Patches were applied and regression testing was conducted from 27 November through 18 December 2006 and documented in reference (c). Review of the vendor's LoC was completed on 29 January 2007. A desktop review was requested to include the following VoIP telephones: i2007 with firmware 0621C4J, i1140E with firmware 0625C4D, i1110 with firmware 0623C4D, and i1120E with firmware 0624C4D. These VoIP telephones were tested with the CS1000E with Software Release 5.0w. The desktop review request was approved on 2 October 2008.

4. The interoperability test summary of the SUT is contained in Table 1. The PBX 1 required and conditional Capability Requirements (CRs) and Feature Requirements (FRs) are listed in Table 2. PBX 2 requirements are a subset of the requirements listed in Table 2. This interoperability test status is based on the SUT's ability to meet:

a. DSN services for Network and Applications specified in reference (d).

b. PBX 1 interface and signaling requirements for trunks/lines specified in reference (e) verified through JITC testing and/or vendor submission of LoC.

c. PBX 1 CRs/FRs specified in reference (e) verified through JITC testing and/or vendor submission of LoC.

d. Internet Protocol version 6 requirements specified in reference (e), paragraph 1.7, Table 1-4, by 30 June 2008 in accordance with reference (f) verified through vendor submission of LoC signed by the Vice President of the company.

e. The overall system interoperability performance derived from test procedures listed in reference (g).

Table 1. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
T1 CAS (DTMF, DP)	Yes	Certified	Met all CRs and FRs with the following minor exceptions: The SUT does not restore the span to service within the required time duration.[1] The SUT recognizes a wink start signal greater than the specified maximum limit.[2] The SUT does not support glare hold resolution for their CAS trunks.[3]
T1 CAS (MFR1)	No	Not Tested	This interface is not supported.[4]
E1 CAS (DTMF, DP)	Yes (Europe only)	Certified	Met all CRs and FRs with the following minor exceptions: The SUT does not restore the span to service within the required time duration.[1] The SUT does not support glare hold resolution for their CAS trunks.[3] The on/off hook pulse that frames the preemption signal on the E1 CAS is intermittently out of the required tolerance of 100 ms (+/-) 5 ms.[5]
E1 CAS (MFR1)	No	Not Tested	This interface is not supported.[4]
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all CRs and FRs with the following minor exception: The SUT fails to automatically return trunks to a maintenance busy condition after the span is broken then restored.[6]
E1 PRI (ITU-T Q.955.3)	No (Europe only)	Certified	Met all CRs and FRs.
DSN Line Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs.
ISDN BRI NI 1/2	No	Certified	Met all CRs and FRs with the following minor exceptions: The SUT does not support NI 2 BRI.[7] The only supported and certified interface is NI 1 BRI with a single appearance of a single directory number.[8] The precedence above ROUTINE ringing cadence that the SUT applies to BRI phones does not meet the specifications.[9] The BRI instruments do not support precedence call waiting.[10]
2-Wire Proprietary Digital	No	Certified	Met all CRs and FRs.
VoIP (ITU-T H.323 with Proprietary Signaling Interface)	No	Certified	Met all CRs and FRs with the following minor exception: Precedence call waiting indication is unique on VoIP phones.[11]
DSN Features and Capabilities			
Features and Capabilities	**Critical**	**Status**	**Remarks**
Common Features	No	Certified	Met all CRs and FRs with the following minor exceptions: The SUT does not correctly support the call forwarding variable feature.[12] The conference disconnect tone that is provided by the SUT does not meet the specifications.[13]
Attendant	No	Certified	Met all CRs and FRs with the following minor exception: Stations cannot be classmarked to prohibit the attendant console from performing a busy override to an active call.[14]
Public Safety	Yes	Certified	Met all CRs and FRs with the following minor exception: The SUT cannot perform a tandem call trace of a specified distant office directory number.[15]
Preset Conferencing	No	Not Tested	This feature is not supported.[16]
Nailed-up Connections	No	Not Tested	This feature is not supported.[16]
Precedence Access Threshold	No	Not Tested	This feature is not supported.[16]
DSN Hotline Services	No	Certified	Met all CRs and FRs with the following minor exception: The SUT does not support a protected hotline specified list.[17]
Network Management	No	Certified	Met all CRs and FRs.
Multiline Hunt Service	No	Certified	Met all CRs and FRs with the following minor exception: The SUT will not permit a BRI station to be a member of a multiline hunt group.[18]
ISDN Services (EKTS)	No	Not Tested	This feature is not supported.[16]

Table 1. SUT Interoperability Test Summary (continued)

DSN Features and Capabilities			
Features and Capabilities	**Critical**	**Status**	**Remarks**
Synchronization	Yes	Certified	Met all CRs and FRs.
Reliability	Yes	Certified	Met all CRs and FRs.
Security	Yes	See note 19.	See note 19.
VoIP System	No	Certified	The SUT is certified for VoIP specifically with certified ASVALAN posted on the JITC TSSI program web page (http://jitc.fhu.disa.mil/tssi/apl.html) approved product list. See note 20.

Network Gateways				
Gateway	**Interface & Signaling**	**Critical**	**Status**	**Remarks**
PSTN	T1 CAS (DTMF, DP)	Yes	Certified	Met all CRs and FRs.
	E1 CAS (DTMF, DP)	No (Europe only)	Certified	Met all CRs and FRs.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all CRs and FRs.
	E1 PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all CRs and FRs.
	Ground Start Line	Yes	Certified	Met all CRs and FRs.
DRSN	TPC 2-Wire analog (GR-506-CORE)	Yes	Certified[21]	Met all CRs and FRs.

LEGEND:

ANSI	American National Standards Institute
ASVALAN	Assured Services Voice Application Local Area Network
BRI	Basic Rate Interface
CAS	Channel Associated Signaling
CFV	Call Forwarding Variable
CRs	Capability Requirements
DISA	Defense Information Systems Agency
DoD	Department of Defense
DP	Dial Pulse
DRSN	Defense Red Switch Network
DSN	Defense Switched Network
DSS1	Digital Subscriber Signaling 1
DTMF	Dual Tone Multi-Frequency
E1	European Basic Multiplex Rate (2.048 Mbps)
EKTS	Electronic Key Telephone System
FRs	Feature Requirements
GR	Generic Requirement
GR-506-CORE	LSSGR: Signaling for Analog Interfaces
GSCR	Generic Switching Center Requirements
H.323	Standard for multi-media communications on packet-based networks
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
ISDN	Integrated Services Digital Network
IT	Information Technology
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
JITC	Joint Interoperability Test Command
LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements
Mbps	Megabits per second
MFR1	Multifrequency Recommendation 1
MLPP	Multi-Level Precedence and Preemption
ms	millisecond
NI 1	National ISDN Standard 1
NI 2	National ISDN Standard 2
NI 1/2	National ISDN Standard 1 or 2
PBX 1	Private Branch Exchange 1
PM	Program Manager
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
Q.931	Signaling Standard for ISDN
Q.955.3	ISDN signaling standard for E1 MLPP
SS7	Signaling System 7
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
T1.607	ISDN – Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
TSSI	Telecom Switched Services Interoperability
TPC	Twisted Pair Copper
VoIP	Voice over Internet Protocol

Table 1. SUT Interoperability Test Summary (continued)

NOTES:

1 When any active trunk interface is physically broken and repaired, the SUT does not restore the span to service and remove the yellow alarm condition within the required time duration. In accordance with the GSCR paragraphs 7.1.4 and 7.2.2, the time required for the removal of the alarm condition after the physical restoration of a broken trunk is 15 (+/-) 5 seconds. The E1 CAS interface can take up to 90 seconds to restore, and all the other interfaces require 30 seconds to be restored. The operational impact is minor since the alarm clears without manual intervention when the span is returned to service.

2 T1 CAS wink start signals greater than the specified maximum limit are recognized as valid by the SUT. The GSCR paragraph 5.3.3.3.1 and GSCR figure 3-2 defines the wink start recognition limits between 100 ms to 350 ms. The SUT recognizes wink start signals from 100 ms to 925 ms in duration. Since all certified switches within the DSN must generate the wink start signal within 140-290 ms, this anomaly has no operational impact.

3 The SUT does not support glare hold resolution on CAS trunks. It only supports glare release. Since glare resolution is conditional for a PBX 1, the operational impact is minor.

4 This interface is not supported. There is no operational impact because it is not a critical requirement.

5 The on/off hook pulse that masks the preemption signal on the E1 CAS is intermittently out of the required tolerance of 100 ms (+/-) 5 ms. The pulse width was measured to be greater than 100 ms about 20 percent of the time with the highest at 128 ms. This never had any impact on the ability of the SUT to support call preemption. Therefore, this anomaly has no operational impact.

6 If a T1 ISDN PRI interface is broken then restored when all channels are in a maintenance busy condition, the SUT fails to automatically return the channels to the previous busy condition. This anomaly has no operational impact because it only occurs when the SUT is in a maintenance condition, and the trunks can be returned to maintenance busy condition manually.

7 The SUT does not support an NI 2 BRI interface. The only supported and certified BRI interface is NI 1. The NI 2 BRI interface is not required for a PBX 1 as specified by GSCR paragraph 2.3.3. The primary differences between NI 1 and NI 2 are supplemental features which currently are not fielded within the DSN nor are there plans to field them in the future. Also, BRI is not required for a PBX 1. This anomaly has a minor operational impact.

8 The SUT will only support a BRI NI 1 voice line with a single directory number and a single appearance of a directory number. However, multiple appearances with different directory numbers can be supported with the digital proprietary instruments which account for the majority of digital instruments fielded within the DSN. Since BRI is not required for a PBX 1, the operational impact is minor.

9 The precedence above ROUTINE ringing cadence that the SUT applies to BRI phones does not meet the specifications as detailed in GSCR 5.5.1 paragraph. The precedence above ROUTINE cadence is distinct from the ROUTINE cadence when it is configured properly; therefore this anomaly has no operational impact.

10 The SUT does not support precedence call waiting for their BRI instruments; however the SUT does support precedence call waiting for all other phone types. Since this is a conditional requirement, there is no operational impact.

11 The SUT supports the "call waiting" indication on VoIP telephones with visual indicators in lieu of audible tones as specified by the GSCR. When call waiting is invoked on a VoIP phone, the phone displays call waiting text along with a flashing symbol. The call waiting symbol flashes twice for a ROUTINE call and three times for precedence above ROUTINE call. Since the requirement for audible tone is conditional, and there are two visual indicators to alert the VoIP user of a waiting call, the operational impact of not supporting audible tones is minor.

12 When call CFV is assigned to any station on the SUT (except BRI, which does not support CFV) and CFV is invoked by the user all precedence calls placed to that instrument are forwarded to the DSN or PSTN. Additionally any station with CFV invoked does not receive a "ping" ring when calls are being forwarded. In accordance with the GSCR, only ROUTINE precedence calls will be forwarded and precedence calls above are diverted to the attendant console, night service, or alternate directory number, therefore this feature is not certified for use within the DSN. This feature is a conditional requirement and will have a minor operational impact.

13 The conference disconnect tone that is provided by the SUT does not meet the specifications designated in GSCR paragraph 5.5.2. The SUT conference disconnect tone is distinguishable from other DSN tones and cadences; therefore this anomaly has a minor operational impact.

14 Stations cannot be classmarked to prohibit the attendant console from performing a busy override to an active call, as specified in the GSCR paragraph 2.2.4. The proper override tone, however, is given to a station active with a call prior to the attendant's bridging into the active call. Since attendants rarely bridge into calls and active calls remain connected when an attendant does bridge into a call, the operational impact is minor.

15 The SUT cannot perform a tandem call trace of a specified distant office directory number as specified in GSCR paragraph 2.4.4. Since this is not required for a PBX 1, this anomaly has a minor operational impact.

16 This feature is not supported. There is no operational impact because it is not a critical requirement.

17 The SUT will not allow the protection of a hotline call originator through the use of a hotline list as required by GSCR paragraph 2.12. However, this capability can be accomplished with the SUT by classmarking authorized hotline users for receiving only calls from other hotline callers. Since this feature is not required by a PBX 1 the operational impact is minor.

18 The SUT will not permit an ISDN BRI station to be a member of a multiline hunt group. All other phone types can be configured as members of a multiline hunt group. This anomaly has a minor operational impact.

19 Security is tested DISA-led Information Assurance test teams and published in a separate report.

20 An IPv6 capable system or product, as defined in the GSCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor Letter of Compliance signed by the Vice President of the company. The vendor must state, in writing, compliance to the following criteria by 30 June 2008**:**

a. Conformant with IPv6 standards profile contained in the DoD IT Standards Registry (DISR).
b. Maintaining interoperability in heterogeneous environments and with IPv4.
c. Commitment to upgrade as the IPv6 standard evolves.
d. Availability of contractor/vendor IPv6 technical support.

21 Interoperability Certification of the SUT does not constitute DRSN PM's approval for connectivity to the DRSN. It is the user's responsibility to request connectivity approval directly from the PM.

Table 2. PBX 1 Requirements

<table>
<tr><th colspan="5">DSN Trunk Interfaces</th></tr>
<tr><th>Interface</th><th>Critical</th><th colspan="2">Requirements
Required or Conditional</th><th>References</th></tr>
<tr><td rowspan="5">T1 CAS
(MFR1, DTMF, DP)

E1 CAS
(MFR1, DTMF, DP)

T1 ISDN PRI NI 1/2
(ANSI T1.619a)

E1 ISDN PRI
(ITU-T Q.955.3)</td><td rowspan="5">No

No
(Europe only)

Yes

No
(Europe only)</td><td>Trunking</td><td>• Framing (R)
• Line Code (R)
• Signaling (R)
• Alarms (R)

• WWNDP (R)
• Outpulsing digit formats (C: CAS only)
• Routing (C)
• Trunk Groups (C)
• Call Processing (C)
• CAS to CCS trunk interworking (C)
• PCM-24/PCM-30 Interoperation (C)
• Direct Inward Dialing (C)</td><td>• GSCR Section 7
• GSCR Section 7
• GSCR Section 5
• GSCR Section 2.5.7, 7.1.4 & 7.2.2
• GSCR Section 4.5.1
• GSCR Section 4.5.2
• GSCR Section 4.2
• GSCR Section 2.5.5 & 2.5.6
• GSCR Section 4
• GSCR Section 3.10
• GSCR Section 7.3
• GSCR Section2.3.2</td></tr>
<tr><td>Voice</td><td>• MOS (R)
• MLPP (R)
• Secure calls (R)</td><td>• CJCSI 6215.01B
• GSCR Section 3
• CJCSI 6215.01B</td></tr>
<tr><td>Facsimile</td><td>• Analog: TIA/EIA-465-A (R)</td><td>• DISR</td></tr>
<tr><td>Data</td><td>• Modem (VBD) (R)
• 56 kbps switched data (R: PRI only)
• 64 kbps switched data (R: PRI only)
• NX56 synchronous BER (R: PRI only)
• NX64 synchronous BER (R: PRI only)
• Secure data (STE/STU-III) (R)</td><td>• CJCSI 6215.01B
• GSCR Section 3.10
• GSCR Section 3.10
• GSCR Section 3.10
• GSCR Section 3.10
• CJCSI 6215.01B</td></tr>
<tr><td>VTC</td><td>• ITU-T H.320 (R: PRI only)</td><td>• DISR</td></tr>
<tr><th colspan="5">DSN Line Interfaces</th></tr>
<tr><td rowspan="5">2-Wire Analog

ISDN BRI NI 1/2
(ANSI T1.619a)

2-Wire Proprietary Digital

VoIP</td><td rowspan="5">Yes

No

No

No</td><td>Access</td><td>• Directory Number Identification (R)
• Line signaling (R)
• Loop Start Line (R: 2-Wire Analog only)
• Alerting Signals and Tones (R)
• WWNDP (R)
• Call Treatments (R)
• 2W user access (R: 2-Wire Analog only)
• Analog busy/idle (R: 2-Wire Analog only)</td><td>• GSCR Section 2.1.1
• GSCR Section 5.2
• GSCR Section 5.2.1
• GSCR Section 5.5
• GSCR Section 4.5
• GSCR Section 4.1
• GSCR Section 4.3.3
• GSCR Section 4.3.4.1</td></tr>
<tr><td>Voice</td><td>• MOS (R)
• Announcements (R)
• MLPP (R)
• Secure Calls (R)</td><td>• CJCSI 6215.01B
• GSCR Section 3.1.3
• GSCR Section 3
• CJCSI 6215.01B</td></tr>
<tr><td>Facsimile</td><td>• Analog: TIA/EIA-465-A (R)</td><td>• DISR</td></tr>
<tr><td>Data</td><td>• Modem (VBD) (R)
• 56 kbps switched data (R: BRI only)
• 64 kbps switched data (R: BRI only)
• NX56 synchronous BER (R: BRI only)
• NX64 synchronous BER (R: BRI only)
• Secure data (STE/STU-III) (R)</td><td>• CJCSI 6215.01B
• GSCR Section 5.7
• GSCR Section 5.7
• GSCR Section 5.7
• GSCR Section 5.7
• CJCSI 6215.01B</td></tr>
<tr><td>VTC</td><td>• ITU-T H.320 (R: BRI only)</td><td>• DISR</td></tr>
</table>

Table 2. PBX 1 Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Common Features	No	• Selective call rejection (C) • Denied originating service (C) • Code restriction and diversion (C) • Call waiting (C) • Three-way calling (C) • Add-on transfer, conference calling, and call hold (C) • Call forwarding (C) • Call pick-up (C)	• GSCR Section 2.1.2 • GSCR Section 2.1.3 • GSCR Section 2.1.4 • GSCR Section 2.1.5 • GSCR Section 2.1.6 • GSCR Section 2.1.7 • GSCR Section 2.1.8 • GSCR Section 2.1.9
Attendant	No	• Initiate all precedence levels (C) • Visual display (C) • Override class of service (C) • Override busy line (C) • Call deflection (C) • Auto recall (C) • Waiting queue (C)	• GSCR Section 2.2.1 • GSCR Section 2.2.2 • GSCR Section 2.2.3 • GSCR Section 2.2.4 • GSCR Section 2.2.5 • GSCR Section 2.2.6 • GSCR Section 2.2.7
Public Safety	No	• Basic Emergency Service (911) (C) • Trace of terminating calls (C) • Outgoing call trace (C) • Tandem call trace (C) • Trace of a call in progress (C)	• GSCR Section 2.4.1 • GSCR Section 2.4.2 • GSCR Section 2.4.3 • GSCR Section 2.4.4 • GSCR Section 2.4.5
Preset Conferencing	No	• Support 10 bridges; 1 originator and 20 conferees per bridge (C) • Assign up to 20 address numbers per bridge (C) • Use KXX codes for bridge access (C) • Conference notification recorded announcement (C) • Auto retrial and alternate address (C) • Bridge release (C) • Lost connection (C) • Secondary conferencing (C) • Address translation (C)	• GSCR Section 2.6 • GSCR Section 2.6 • GSCR Section 2.6 • GSCR Section 2.6.1 • GSCR Section 2.6.2 • GSCR Section 2.6.3 • GSCR Section 2.6.4 • GSCR Section 2.6.5 • GSCR Section 2.7
Nailed-up Connections	No	• Between any two like terminations (C) • PCM-24 and PCM-30, both CAS and CCS (C) • Supervision passed end-to-end for A/D or D/A (C) • Monitored and auto reconfigure (C) • Support at least 10% of circuits as nailed-up (C) • Non-preemptable (C)	• GSCR Section 2.8 • GSCR Section 2.8 • GSCR Section 2.8 • GSCR Section 2.8 • GSCR Section 2.8 • GSCR Section 2.8
PAT	No	• Classmark for/not for PAT screening (C) • 7 PAT mechanisms (C) • Outgoing call screening (C) • Functional structure (C) • Simultaneous calls limitation (C) • Overflow process (C) • Decrementing call-in-progress count (C) • Call treatment (C) • Queuing (C) • Attendant calls (C) • Operations measurement registers (C) • Maintenance and Administration of thresholds (C)	• GSCR Section 2.11.1 • GSCR Section 2.11.1 • GSCR Section 2.11.1.1 • GSCR Section 2.11.1.2 • GSCR Section 2.11.1.3 • GSCR Section 2.11.1.4 • GSCR Section 2.11.1.5 • GSCR Section 2.11.1.6 • GSCR Section 2.11.1.7 • GSCR Section 2.11.1.8 • GSCR Section 2.11.1.9 • GSCR Section 2.11.1.10
DSN Hotline Services	No	• Hotline restrictions (C) • Auto initiate (C) • Analog and digital (C) • Subscription basis (C) • Protected hotline calling (C) • WWNDP interoperable (C)	• GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12.1-4 • GSCR Section 2.12.5

Table 2. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Network Management	No	• Interfaces (C) • Measurements and data generation (C) • Fault management (C) • Configuration management (C) • Accounting management (C) • Performance management (C) • NM controls (C) • Remote access (C)	• GSCR Section 9.1 • GSCR Section 9.2 • GSCR Section 9.3 • GSCR Section 9.4 • GSCR Section 9.5 • GSCR Section 9.6 • GSCR Section 9.7 • GSCR Section 9.8
ISDN Services	No	• Electronic Key Telephone Systems (EKTS) (C)	• GSCR Section 10, table 10-3
Synchronization	Yes	• Line timing mode (R) • Internal Stratum 4 (R)	• GSCR Section 11.1.1.2 • GSCR Section 11.1.2.2
Reliability	Yes	• GR-512-CORE (R)	• GSCR Section 12
Security	Yes	• GR-815, STIGs, and DIACAP (replacement for DITSCAP) (R)	• GSCR Section 13
VoIP			
VoIP System	No	VoIP function is conditional. If VoIP is provided, **all** of the following requirements must be met: • Voice Quality with MOS of 4.0 or better • Class of Service (CoS) and Quality of Service (QoS) • ITU-T G.711 PCM Codec • Traffic Engineering • Security • NM • Line timing • Internal Clock • Latency ≤ 60 milliseconds • Packet Loss • IPv6 capable	• GSCR Appendix 3 • GSCR Appendix 3 • GSCR Appendix 3 • GSCR Appendix 3 • GSCR Appendix 3 • GSCR Appendix 3 • GSCR Appendix 3 • GSCR Appendix 3 • GSCR Appendix 3 • GSCR Appendix 3 • GSCR Appendix 3, paragraph 1.7

Table 2. PBX 1 Requirements (continued)

Network Gateways				
Gateway	**Critical**	**Requirements Required or Conditional**		**References**
PSTN[1]	No	Trunking	• Positive Identification Control (C) • On-Netting (C) • Off-Netting (C)	• CJCSI 6215.01B • CJCSI 6215.01B • CJCSI 6215.01B
DRSN[2]	Yes	Access	• Alerting Signals and Tones (R) • Call Processing (R) • Call Treatments (R) • Analog busy/idle (R)	• GSCR Section 5.5 • GSCR Section 4.4 • GSCR Section 4.1 • GSCR Section 4.3.4.1
		Voice	• MOS (C) • MLPP (C) • Secure calls (C)	• CJCSI 6215.01B • GSCR Section 3 • CJCSI 6215.01B

LEGEND:

2W - 2-Wire
A/D - Analog to Digital Conversion
ANSI - American National Standards Institute
BER - Bit Error Ratio
BRI - Basic Rate Interface
C - Conditional
CAS - Channel Associated Signaling
CCS - Common Channel Signaling
CJCS - Chairman of the Joint Chiefs of Staff
CJCSI - CJCS Instruction
D/A - Digital to Analog Conversion
DIACAP - DoD Information Assurance Certification and Accreditation Process
DISA - Defense Information Systems Agency
DISR - DoD IT Standards Registry
DITSCAP - DoD IT Security Certification and Accreditation Process
DoD - Department of Defense
DP - Dial Pulse
DRSN - Defense Red Switch Network
DSN - Defense Switched Network
DTMF - Dual Tone Multi-Frequency
E1 - European Basic Multiplex Rate (2.048 Mbps)
EIA - Electronic Industries Alliance
G.711 - Standard for PCM of Voice Frequencies
GR - Generic Requirement
GR-512 - LSSGR: Reliability, Section 12
GR-815 - Generic Requirements For Network Element/Network System (NE/NS) Security
GSCR - Generic Switching Center Requirements
H.320 - Standard for Narrowband VTC
IPv6 - Internet Protocol version 6
ISDN - Integrated Services Digital Network
IT - Information Technology
ITU-T - International Telecommunication Union-Telecommunication Standardization Sector
kbps - kilobits per second
KXX - K= any number 2-8; X= any number 1-9
LAN - Local Area Network
Mbps - Megabits per second
MFR1 - Multi-Frequency Recommendation 1
MLPP - Multi-Level Precedence and Preemption
MOS - Mean Opinion Score
NI 1/2 - National ISDN Standard 1 or 2
NM - Network Management
NX56 - Data format restricted to multiples of 56 kbps
NX64 - Data format restricted to multiples of 64 kbps
PAT - Precedence Access Threshold
PBX 1 - Private Branch Exchange 1
PCM - Pulse Code Modulation
PCM-24 - Pulse Code Modulation - 24 Channels
PCM-30 - Pulse Code Modulation - 30 Channels
PRI - Primary Rate Interface
PSTN - Public Switched Telephone Network
Q.955.3 - ISDN Signaling Standard for E1 MLPP
R - Required
SMEO - Small End Office
SS7 - Signaling System 7
STE - Secure Terminal Equipment
STIGs - Security Technical Implementation Guides
STU-III - Secure Telephone Unit - 3rd generation
T1 - Digital Transmission Link Level 1 (1.544 Mbps)
T1.619a - SS7 and ISDN MLPP Signaling Standard for T1
TIA - Telecommunications Industry Association
TIA/EIA-465-A - Group 3 Facsimile Apparatus for Document Transmission
VBD - Variable bit data
VoIP - Voice over Internet Protocol
VTC - Video Teleconferencing
WWNDP - Worldwide Numbering and Dialing Plan

NOTES:

1 Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.
2 Facsimile, data, and VTC services are not provided via the DSN to DRSN interface.

5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet), or http://199.208.204.125 (SIPRNet). Information related to DSN testing is on the TSSI website at http://jitc.fhu.disa.mil/tssi.

6. The JITC point of contact is Captain Oskar Widecki, DSN 879-5269, commercial (520) 538-5269, FAX DSN 879-4347, or e-mail to Oskar.Widecki@disa.mil. The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The tracking number for the SUT is 51825.

FOR THE COMMANDER:

for J. T. Schutte

Enclosure a/s

RICHARD A. MEADOR
Chief
Battlespace Communications Portfolio

Distribution (electronic mail):
Joint Staff J-6
Joint Interoperability Test Command, Liaison, TE3/JT1
Office of Chief of Naval Operations, CNO N6F2
Headquarters U.S. Air Force, Office of Warfighting Integration & CIO, AF/XCIN (A6N)
Department of the Army, Office of the Secretary of the Army, DA-OSA CIO/G-6 ASA (ALT), SAIS-IOQ
U.S. Marine Corps MARCORSYSCOM, SIAT, MJI Division I
DOT&E, Net-Centric Systems and Naval Warfare
U.S. Coast Guard, CG-64
Defense Intelligence Agency
National Security Agency, DT
Defense Information Systems Agency, TEMC
Office of Assistant Secretary of Defense (NII)/DOD CIO
U.S. Joint Forces Command, Net-Centric Integration, Communication, and Capabilities Division, J68
Defense Information Systems Agency, GS23

ADDITIONAL REFERENCES

(c) Joint Interoperability Test Command, Memo, JTE, "Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS) 1000M Cabinet and CS1000M Chassis (including Voice over Internet Protocol [VoIP]) and DSN Option 11C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages," 7 March 2007

(d) Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01B, "Policy for Department of Defense Voice Services," 23 September 2001

(e) Defense Information Systems Agency, "Department of Defense Voice Networks Generic Switching Center Requirements (GSCR), Incorporated Change 1," 1 March 2005

(f) Executive Office of the President, "Transition Planning for Internet Protocol version 6 (IPv6)," 2 August 2005

(g) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 1, Revision 1," 1 June 2005